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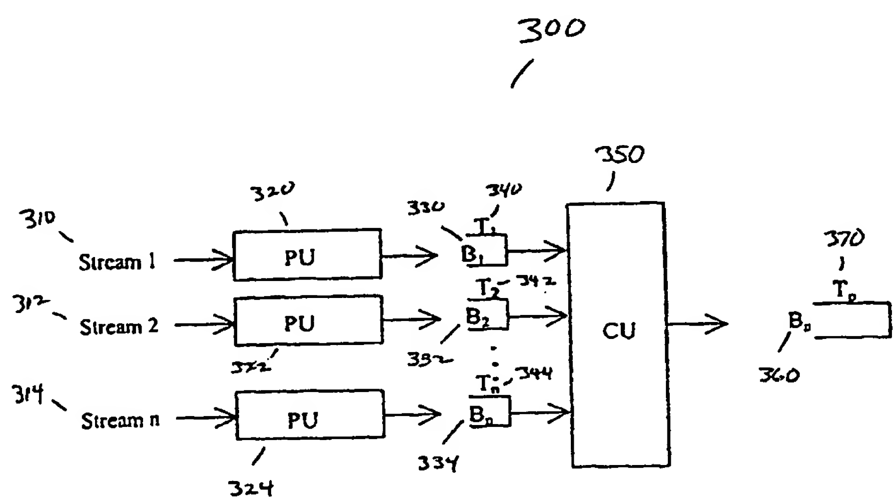



(57) Abstract

A flexible mechanism employing timers within the gateways (300) to facilitate efficient multiplexing. Timers (340, 342, 344) are provided, wherein their value can be adjusted on a dynamic basis depending on factors such as network congestion that impact end-to-end delay. The extraction of data from the buffers (330, 332, 334) is triggered when either the timer (370) expires or when the accumulated data reaches a certain size. There are two ways one could set the timer values. In the first case, the network operator chooses the value based on the known approximation of end-to-end delay. In the second case, the multiplexing controller (350) has the capability to extract the network delay information from the RTCP reports.

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# METHOD AND APPARATUS FOR PROVIDING EFFICIENT MULTIPLEXING BETWEEN GATEWAYS USING DYNAMIC TIMERS

## BACKGROUND OF THE INVENTION

### Field of the Invention.

This invention relates in general to a IP telephone, and more particularly to a method and apparatus for providing efficient multiplexing between gateways using dynamic timers

### Description of Related Art.

Public switched telephone networks (PSTN) provide users with dedicated, end-to-end circuit connections for the duration of each call. Circuits are reserved between the originating switch, tandem switches (if any), and the terminating switch based on the called party number. The PSTN also provides access to intelligent network services using the Signaling System 7 (SS7) protocol. SS7 is used for basic call setup, management, and tear down and to query databases that support intelligent network services such as local number portability, mobile subscriber authentication and roaming, virtual private networking, and toll-free (800) service.

In contrast to circuit-switched PSTN, packet-switched Internet Protocol (IP) networks provide shared, virtual circuit connections between users. Bandwidth is dynamically allocated for improved utilization of network capacity. IP packets are routed to the destination IP address contained within the header of each packet. Packets may travel over separate network paths before arriving at their final destination for reassembly and resequencing. The transmission speed between any two users can change dramatically based on the dynamic number of users sharing the common transmission medium, their bandwidth requirements, the capacity of the transmission medium, and the efficiency of the network routing and design.

The IP is a network layer protocol that routes data across an Internet. The Internet Protocol was designed to accommodate the use of host and routers built by different vendors, encompass a growing variety of growing network types, enable the network to grow without interrupting servers, and support higher-layer of session and message-oriented services. The IP network layer allows integration of Local Area Network "islands".

The Internet not only provides a medium for data transport, but also has developed as a medium for telecommunications. In fact, IP telephony is maturing as a technology and a service, which places it in direct conflict with the conventional public telephone network. New types of IP telephony equipment are being introduced and small Internet service providers and niche carriers are beginning to offer IP telephony services.

However, while IP telephony, or voice-over-IP, has great potential to compete against the traditional public voice network, many obstacles must first be overcome. For example, the quality of Internet telephone calls is not as good as public network calls. Customer interest in the United States-where long-distance prices have consistently fallen-is uncertain. Further, IP telephony network equipment is new and lacks features that carriers desire.

Consequently, only a handful of small telephone companies now offer IP telephony services, and the amount of traffic they carry is minimal. However, equipment vendors and carriers are moving quickly to overcome these problems and, according to analysts, the number of established carriers beginning IP telephony trials is rising rapidly, new carriers are jumping into the market and the number of services being offered is expected to increase dramatically. In fact, analysts predict that users will embrace the new IP-based voice services.

Accordingly, the success of Internet has further consolidated the notion that IP is the dominant transport protocol in access networks. The penetration of Internet and subsequent IP connectivity to homes and businesses have given impetus to integrated services over IP which means voice, data and video over a single IP network. At present, IP networks offer a single class of service called best effort, which can not guarantee any Quality of Service (QoS) to applications. To support delay sensitive applications such as voice and interactive multimedia, there have been many proposals submitted to the Internet Engineering Task Force (IETF) on how to integrate QoS in IP networks. These proposals include differentiated service (diff-serv), Integrated services (Int-serv) and Multi Protocol Label Switching (MPLS). Despite these efforts, QoS in IP is still elusive and could take some time before it is deployed over global Internet.

As mentioned above, IP telephone has emerged as a potential application to challenge the traditional phone companies by offering long distance telephone service over Internet for low prices. There are a large number of equipment vendors offering IP telephone gateways and accessories to provide IP telephony service to corporate customers and Internet Service Providers (ISPs). IP telephone standards

such as H.323, H.225 and H.245 have been standardized to enhance the rapid deployment of IP telephone services in global Internet. Even though, IP telephone is not a reality in public Internet today, it has been successful in Intranet and Virtual Private Networks (VPN) environments.

As described above, voice is carried in a circuit switched network with connection oriented model. The explosive growth of data traffic in the network has changed the notion of service specific networks (single network for single service). Voice over IP network enables the voice traffic from public branch exchanges (PBX) and public switched telephone network (PSTN) users to share the data network thus improving the network utilization and lowering the cost associated with long distance telephone network.

IP telephone gateways act as an interface between the existing PSTN and PBX networks and IP networks. This method allows one PSTN user to call another PSTN user connected through IP telephone gateways thus eliminating the need for long distance telephone network.

In a IP telephony connection, two sides of the PSTN/PBX users (two branches of the same company) are interconnected by IP telephone gateways. In such application, a telephone call between PSTN/PBX users located at either side of the gateways is carried by a separate Real-time Transport Protocol/User Datagram Protocol/Internet Protocol (RTP/UDP/IP) connection. RTP is an Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol, although the specification is general enough to support other transport protocols. The User Datagram Protocol is a connectionless protocol that, like TCP, runs on top of IP networks. Unlike TCP/IP, UDP/IP provides very few error recovery services, offering instead a direct way to send and receive datagrams over an IP network.

IP telephony gateways provide an interface between the existing circuit switched telephone networks (such as PSTN and GSM) and the packet switched IP data networks. In traditional IP telephony applications, telephone calls between PSTN users interconnected by a pair of IP telephony gateways to compress incoming PSTN speech samples generate packets with sizes ranging from 5 to 20 bytes per speech sample.

For example, G.723.1 (the most popular IP telephony codec and the International Multimedia Teleconferencing Consortium's (IMTC) Voice over IP (VoIP)

mandatory low bit-rate codec), generates a 20 byte speech packet at 30 ms intervals. Many codecs used in cellular environment generate less than 10 byte packet per speech sample. Small size packets are subjected to large overhead when transferred using the Real time Transport Protocol (RTP). The RTP/UDP/IP overhead is 40 bytes (12+8+20) for a simple speech packet. For example, a 10 byte packet transferred via RTP/UDP/IP increases the overhead to 80% (40 byte overhead/50 byte overhead plus packet). In addition, for each call request a single UDP/IP connection (a pair of UDP ports) is established between the gateways requiring a large state (memory) to be maintained at the telephony gateways, thereby making these less scaleable.

When users on a circuit switched voice network (such as PSTN or GSM) communicate by traversing a packet switched network (such as an IP network), a PSTN-IP or GSM-IP (as the case may be) gateway is needed at the network boundaries. In such a case, it is advantageous to multiplex several users (or voice calls) traversing the same pair of telephone gateways.

In a user-multiplexing scenario, the main goal is to improve the bandwidth efficiency by reducing overhead per packet. However, user multiplexing needs a set of timers so that the delay bound for packet streams is satisfied. In IP networks, network delay (propagation and transmission delay) is a time varying quantity. At present, there is no mechanism to guarantee a fixed delay between point A and point B in IP networks. Real Time Control Protocol (RTCP) provides a receiver generated reports at periodic intervals that include network delay in forward direction for a packet stream.

It can be seen then that there is a need for a flexible mechanism employing timers within the gateways to facilitate efficient multiplexing.

It can also be seen that there is a need for a mechanism to adjust the timer values based on the RTCP reports so that multiplexing efficiency can be improved without violating the delay bound.

It can also be seen that there is a need for a method that allows the values for then timers to be set dynamically based on the network delay observed during the multiplexing process.

## SUMMARY OF THE INVENTION

To overcome the limitations in the prior art described above, and to overcome other limitations that will become apparent upon reading and understanding the present specification, the present invention discloses a flexible mechanism employing timers within the gateways to facilitate efficient multiplexing.

The present invention solves the above-described problems by providing timers, wherein their value can be adjusted on a dynamic basis depending on factors such as network congestion that impact end-to-end delay. The extraction of data from the buffers is triggered when either the timer expires or when the accumulated data reaches a certain size. There are two ways one could set the timer values. In the first case, the network operator chooses the value based on the known approximation of end-to-end delay. In the second case, the multiplexing controller has the capability to extract the network delay information from the RTCP reports.

A method in accordance with the principles of the present invention includes receiving a stream of data packets on N channels, storing the packets for the Nth channel in an Nth input buffer having a timer associated therewith, moving the packets in the Nth input buffer to a packetization buffer having a timer associated therewith in response to the timer associated with the Nth input buffer, and transmitting the packets in the packetization buffer in response to the timer associated with the packetization buffer.

Other embodiments of a method in accordance with the principles of the invention may include alternative or optional additional aspects. One such aspect of the present invention is that the timer associated with the packetization buffer is set to meet a first set of criteria and to meet an end-to-end delay associated with the packets being transmitted.

Another aspect of the present invention is that the timer associated with the Nth input buffer is set based on the setting of the timer associated with the packetization buffer and the end-to-end delay.

Another aspect of the present invention is that the first set of criteria comprises maximizing the time the packets are held in the Nth input buffer, moving the packets in the Nth input buffer to the packetization buffer according to a first threshold, and transmitting the packets in the packetization buffer according to a second threshold.

Another aspect of the present invention is that the first threshold comprises expiration of the timer associated with the Nth input buffer.

Another aspect of the present invention is that the timer associated with the Nth input buffer is set at a value that expires before the Nth input buffer overflows.

Another aspect of the present invention is that the first threshold comprises detection of an overflow condition for the Nth input buffer.

Another aspect of the present invention is that the second threshold comprises expiration of the timer associated with the Nth input buffer.

Another aspect of the present invention is that the second threshold comprises a maximum packet payload size, the packets in the packetization buffer being transmitted when the packets in the packetization buffer exceeds the maximum packet payload size.

These and various other advantages and features of novelty which characterize the invention are pointed out with particularity in the claims annexed hereto and form a part hereof. However, for a better understanding of the invention, its advantages, and the objects obtained by its use, reference should be made to the drawings which form a further part hereof, and to accompanying descriptive matter, in which there are illustrated and described specific examples of an apparatus in accordance with the invention.

## BRIEF DESCRIPTION OF THE DRAWINGS

Referring now to the drawings in which like reference numbers represent corresponding parts throughout:

Fig. 1 illustrates one possible embodiment of an advanced communication network ;

Fig. 2 illustrates a system block diagram showing the use of gateways between a circuit-switched network (CSN) and a packet-switched network (PSN); and

Fig. 3 illustrates a PSTN-IP/Mobile-IP gateway depicting the various buffers and timers according to the present invention.

## DETAILED DESCRIPTION OF THE INVENTION

In the following description of the exemplary embodiment, reference is made to the accompanying drawings which form a part hereof, and in which is shown by way of illustration the specific embodiment in which the invention may be practiced. It is to be understood that other embodiments may be utilized as structural changes may be made without departing from the scope of the present invention.

The present invention provides a flexible mechanism employing timers within the gateways to facilitate efficient multiplexing. The timers include values that can be adjusted on a dynamic basis depending on factors such as network congestion that impact end-to-end delay. The extraction of data from the buffers is triggered when either the timer expires or when the accumulated data reaches a certain size. There are two ways one could set the timer values. In the first case, the network operator chooses the value based on the known approximation of end-to-end delay. In the second case, the multiplexing controller has the capability to extract the network delay information from the RTCP reports.

Fig. 1 illustrates one possible embodiment of an advanced communication network 100, e.g., a third generation GSM evolution. Those skilled in the art will recognize that the present invention is not meant to be limited to use with GSM mobile communication systems, but is applicable to other mobile communication systems. However, the present invention will be described herein using GSM as an example.

As shown in Fig. 1, the first implementations of Generic Radio Access Network (GRAN) may be based on the integration of RAN and SGM/UMTS core network, which has been evolved from the GSM core network by integrating new third generation capabilities. The evolved GSM network elements are referred to as 3G MSC and 3G SGSN.

A mobile unit 110 receives and sends signals to a base station (BS) 112. Base stations 112 are in turn coupled to a radio network controller (RNC) 114 in the radio access network (RAN) 116. The RAN interfaces with GSM/UMTS core network 120 via the lu-interface 122, which corresponds to the GSM A-interface and GPRS Gb-interface. As can be seen, radio access 130 is isolated from the core network 120, and the goal is that the GSM/UMTS core network would have the flexibility to support any radio access scheme. Circuit switched services are routed via the GSM MSC 140, and the packet switched services via the GPRS part 150 of the GSM/UMTS core network.

Fig. 2 illustrates a system block diagram 200 showing the use of gateways 210, 212 between a circuit-switched network (CSN) 220 and a packet-switched network (PSN) 230. Multiple users traversing the same pair of gateways 210, 212 are multiplexed between the two gateways 210, 212 for traversal over the PSN 230. Multiplexing voice streams 240 between IP telephony gateways 210, 212 needs to be done in an efficient manner while not compromising the user's performance requirements.

Fig. 3 illustrates a PSTN-IP/Mobile-IP gateway 300 depicting the various buffers and timers according to the present invention. In Fig. 3, multiple streams 310, 312, 314 are received at corresponding processing units (PU) 320, 322, 324. Buffers 330, 332, 334 are associated with each voice stream 310, 312, 314. Timers 340, 342, 344 are associated with each buffer 330, 332, 334 for providing efficient multiplexing between gateways 210, 212 as shown in Fig. 2. The buffers 330, 332, 334 provide packets to the multiplexing controller unit (CU) 350. the multiplexing controller unit 350 then packetizes the voice streams 310, 312, 314 at a packetization buffer 360 having a packetization timer 370 associated therewith.

The times 340, 342, 344 of the PSTN-IP/Mobile-IP gateway 300 provide efficiency in multiplexing voice steams between such IP gateways. According to the present invention, there are two ways for setting up the values of timers 340-344, 370. First, the network operator may choose the value based on the known approximation of end-to-end delay. Alternatively, the multiplexing controller 350 has the capability to extract the network delay information from Real-Time Control Protocol (RTCP) reports.

In Fig. 3, the gateway functionality is not illustrated in its entirety, but is drawn merely for illustration purposes. The buffer organization could be arbitrary, so that the buffers could either be physically separate buffers or logical buffers that use some dynamic sharing principle. Each incoming voice stream $x$ 310-314 is processed by a processing unit 320-324 prior to being stored in buffer $B_x$ 330-334. Such processing could include transcoding and encryption.

Associated with each voice stream buffer or input buffer $B_x$ 330-334 is a timer $T_x$ 340-344, while a timer $T_p$ 370 is associated with the packetization buffer $B_p$ 360. Based on the occupancy of each voice stream buffer 330-334 and the timer values 340-344, the control unit (CU) 350 moves voice samples from the voice stream buffers 330-334 to the packetization buffer 360. The control unit 350 prepends a suitable header to the voice sample(s) taken from the voice stream buffers 330-334 prior to storing it in the packetization buffer 360.

Upon transferring data from a voice stream buffers $B_x$ 330-334 to the packetization buffer 360, the voice stream timers $T_x$ 340-344 are reinitialized and the packetization timer $T_p$ 370 is suitably adjusted. Based on the value of the timer $T_p$ 370 and on the occupancy of the packetization buffer 360, data from the packetization buffer 360 is sent out into the network after suitably prepending it with the necessary headers. Upon doing so, the timer $T_p$ 370 is re-initialized.

Referring to Fig. 3, the total end-to-end delay $D$ experienced by a packet equals the sum of the delay on the circuit switched network ($d_c$) 250, the delay within the voice stream buffer ($d_{vb}$) 252, the delay within the packetization buffer ($d_{pb}$) 254, other processing delays within the initiating gateway ($d_{pl}$) 256, transmission delay from the initiating gateway onto the network ($d_t$) 258, propagation delay over the packet switched network ($d_p$) 260, and all processing delays in the remote gateway ($d_{p2}$) 262.

$$D = d_c + d_{vb} + d_{pb} + d_{pl} + d_t + d_p + d_{p2}$$

The delay over the circuit switched network $d_c$ 250 can be determined and is relatively unchanging. Processing delays at the gateways while depending on the load at the gateways, do not vary significantly. Hence, $d_{pl}$ 256 and $d_{p2}$ 258 are relatively constant and can be determined. The transmission delay $d_t$ 258 depends on the packet size and the link speed, and the propagation delay $d_p$ 260 depends on several factors including the congestion in the network. Feedback about congestion in the network can be obtained from several mechanisms including RTCP (Real Time Control Protocol) reports, thus giving a handle on $d_p$ 260. Thus, a knowledge of all parameters contributing to the end-to-end delay of a call, except the two buffer delays $d_{vb}$ 252 and $d_{pb}$, 254 is available. Given the delay $D$ that is tolerable by the user, the sum of the two packetization delays is known. Thus, we have

$$d_{vb} + d_{pb} = c,$$

where $c$ is some known (potentially time varying) value. Our goal is to select the value for "$c$" such that end-to-end delay is either less than or equal to D. Under this assumption it can be shown that

$$c = f(d_n),$$

where $d_n = d_p + d_1$ and $d_n$ is the network delay, $d_p$ is the Propagation delay, $d_1$ is the Transmission delay, and $f(.)$ denotes some function.

The network delay value $d_n$ is extracted from the RTCP reports and used to set the value for $c$ (in essence values for $d_{vb}$ 252 and $d_{pb}$ 254).

Several schemes for initiating the timer values and for moving data out of a buffer are possible. Referring to Fig. 3, the present invention is implementing using the following general recommendations:

- Voice samples should be held in the voice buffers 330-334 for as long as possible prior to moving them to the packetization buffer 360. A larger amount of data (i.e., several voice samples from the same source) when moved from the voice buffers 330-334 to the packetization buffer 360 incurs less overhead due to a single header being prepended to a larger amount of data.
- Voice samples are moved from the voice buffers 330-334 to the packetization buffer 360 as soon as the corresponding timers 340-344 expire or when the amount of data in the voice buffers 330-334 is such that a new incoming sample

would result in the data in the voice buffers 330-334 exceeding that which can be handled by the length indicator in the prepended header.

- At any given time, voice samples placed in a voice buffers 330-334 should not exceed the buffer size. The buffer timers 340-344 should be set such that the voice samples are removed before an overflow occurs.
- Data from the packetization buffer 360 is transmitted onto the network as soon as the timer $T_p$ 370 expires or when the amount of data in the packetization buffer 360 just exceeds the maximum packet payload size. The new incoming data from the voice buffers 330-334 that resulted in the packet payload size exceeding the limit is transmitted in the next packet. The maximum packet payload size is governed by the MTU (maximum transmission unit) within the network. This is usually 1500 bytes for IP datagrams over the public Internet.

The common packetization buffer timer $T_p$ 370 is established with the end-to-end delay $D_x$ of each voice stream $x$ and the above specified guidelines in mind. Having established the value that $T_p$ 370 needs to be initialized with, the value that each voice buffer timer $T_x$ 370 is initialized with is ascertained merely by subtracting $T_p$ 370 from the known constant $c$ associated with the particular voice stream $x$ 310-314.

When the first voice sample for voice stream $x$ 310-314 is stored in the corresponding voice stream buffer $B_x$ 330-334, the timer Tx 340-344 associated with the voice stream buffer 330-334 is re-initialized. When the timer $T_x$ 340-344 expires or when the length of samples in the voice buffers 330-334 reaches the maximum length that can be handled by the header (due to a constraint on the length indicator within the header), then the samples are transferred from the voice buffers 330-334 to the packetization buffer 360 after prepending the suitable header. When data is transferred from the voice buffers 330-334 to the packetization buffer 360, the timer associated with the packetization buffer $T_p$ 370 is adjusted to $T_p = min(T_x, T_p)$, where $min()$ is the function that chooses the minimum of the two values $T_x$ and $T_p$. The timers $T_x$ 340-344 associated with the voice buffers $B_x$ 330-334 are is then re-initialized. The data from the packetization buffer 360 is sent out to the network when the timer $T_p$ 370 expires or when the amount of data within the packetization buffer 360 matches the MTU (taking into account packetization headers). Upon transmission of a packet from the packetization buffer 360 and subsequently upon the arrival of the first data from any voice stream buffer 330-334 to the packetization buffer 360, the timer $T_p$ 370 is re-initialized.

The foregoing description of the exemplary embodiment of the invention has been presented for the purposes of illustration and description. It is not intended to be exhaustive or to limit the invention to the precise form disclosed. Many modifications and variations are possible in light of the above teaching. It is intended that the scope of the invention be limited not with this detailed description, but rather by the claims appended hereto.

WHAT IS CLAIMED IS:

1. A method for providing efficient multiplexing between gateways, comprising:

receiving a stream of data packets on a first channel;

storing the packets in a input buffer having a timer associated therewith;

moving the packets in the input buffer to an packetization buffer having a timer associated therewith in response to the timer associated with the input buffer;

transmitting the packets in the packetization buffer in response to the timer associated with the packetization buffer.

2. The method of claim 1 wherein the timer associated with the packetization buffer is set to meet a first set of criteria and to meet an end-to-end delay associated with the packets being transmitted.

3. The method of claim 2 wherein the timer associated with the input buffer is set based on the setting of the timer associated with the packetization buffer and the end-to-end delay.

4. The method of claim 2 wherein the first set of criteria comprises maximizing the time the packets are held in the input buffer, moving the packets in the input buffer to the packetization buffer according to a first threshold, and transmitting the packets in the packetization buffer according to a second threshold.

5. The method of claim 4 wherein the first threshold comprises expiration of the timer associated with the input buffer.

6. The method of claim 5 wherein the timer associated with the input buffer is set at a value that expires before the input buffer overflows.

7. The method of claim 4 wherein the first threshold comprises detection of an overflow condition for the input buffer.

8. The method of claim 4 wherein the second threshold comprises expiration of the timer associated with the input buffer.

9. The method of claim 4 wherein the second threshold comprises a maximum packet payload size, the packets in the packetization buffer being transmitted when the packets in the packetization buffer exceeds the maximum packet payload size.

10. A method for providing efficient multiplexing between gateways, comprising:

storing voice samples for an xth voice stream in a corresponding xth voice stream buffer;

re-initializing a xth timer associated with the xth voice stream buffer;

pre-pending a header to the samples in the xth voice stream when the xth timer expires or when the samples in the xth voice buffer reaches a maximum length that can be handled by the header;

transferring the samples from the xth voice buffer to a packetization buffer after prepending the header;

adjusting a timer associated with the packetization buffer to a minimum value, the minimum being the least of a value for the timer associated with the xth voice stream buffer or a value for the timer associated with the packetization buffer;

re-initializing the timer associated with the xth voice stream buffer; and

transmitting the samples in the packetization buffer when the timer associated with the packetization buffer expires or when the samples in the packetization buffer exceed a maximum packet payload size.

11. A method for providing efficient multiplexing between gateways, comprising:

receiving a stream of data packets on N channels;

storing the packets for the Nth channel in an Nth input buffer having a timer associated therewith;

moving the packets in the Nth input buffer to a packetization buffer having a timer associated therewith in response to the timer associated with the Nth input buffer;

transmitting the packets in the packetization buffer in response to the timer associated with the packetization buffer..

12. The method of claim 11 wherein the timer associated with the packetization buffer is set to meet a first set of criteria and to meet an end-to-end delay associated with the packets being transmitted.

13. The method of claim 12 wherein the timer associated with the Nth input buffer is set based on the setting of the timer associated with the packetization buffer and the end-to-end delay.

14. The method of claim 12 wherein the first set of criteria comprises maximizing the time the packets are held in the Nth input buffer, moving the packets in the Nth input buffer to the packetization buffer according to a first threshold, and transmitting the packets in the packetization buffer according to a second threshold.

15. The method of claim 14 wherein the first threshold comprises expiration of the timer associated with the Nth input buffer.

16. The method of claim 15 wherein the timer associated with the Nth input buffer is set at a value that expires before the Nth input buffer overflows.

17. The method of claim 14 wherein the first threshold comprises detection of an overflow condition for the Nth input buffer.

18. The method of claim 14 wherein the second threshold comprises expiration of the timer associated with the Nth input buffer.

19. The method of claim 14 wherein the second threshold comprises a maximum packet payload size, the packets in the packetization buffer being

transmitted when the packets in the packetization buffer exceeds the maximum packet payload size.

20. A gateway, comprising:

a plurality of input buffers, each of the plurality of input buffers receiving samples from an input stream;

a timer associated with each of the plurality of input buffers;

a packetization buffer for storing samples from the plurality of input buffers before transmission over a network;

a timer associated with the packetization buffer; and

a control unit for moving samples from the plurality of input buffers to the packetization buffer in response to the timers associated with the plurality of input buffers and for prepending a suitable header to the samples taken from the plurality of input buffers prior to storing the samples in the packetization buffer, the controller transmitting the samples in the packetization buffer in response to the timer associated with the packetization buffer.

21. The gateway of claim 20 wherein the timer associated with the packetization buffer is set to meet a first set of criteria and to meet an end-to-end delay associated with the samples being transmitted.

22. The gateway of claim 21 wherein the timer associated with an input buffer is set based on the setting of the timer associated with the packetization buffer and the end-to-end delay.

23. The gateway of claim 21 wherein the first set of criteria comprises maximizing the time the samples are held in the plurality of input buffers, moving the samples in the plurality of input buffers to the packetization buffer according to a first threshold, and transmitting the samples in the packetization buffer according to a second threshold.

24. The gateway of claim 23 wherein the first threshold comprises expiration of the timer associated with an input buffer.

25. The gateway of claim 24 wherein the timer associated with an input buffer is set at a value that expires before the input buffer overflows.

26. The gateway of claim 23 wherein the first threshold comprises detection of an overflow condition for an input buffer.

27. The gateway of claim 23 wherein the second threshold comprises expiration of a timer associated with am input buffer.

28. The gateway of claim 23 wherein the second threshold comprises a maximum packet payload size, the samples in the packetization buffer being transmitted when the samples in the packetization buffer exceeds the maximum packet payload size.

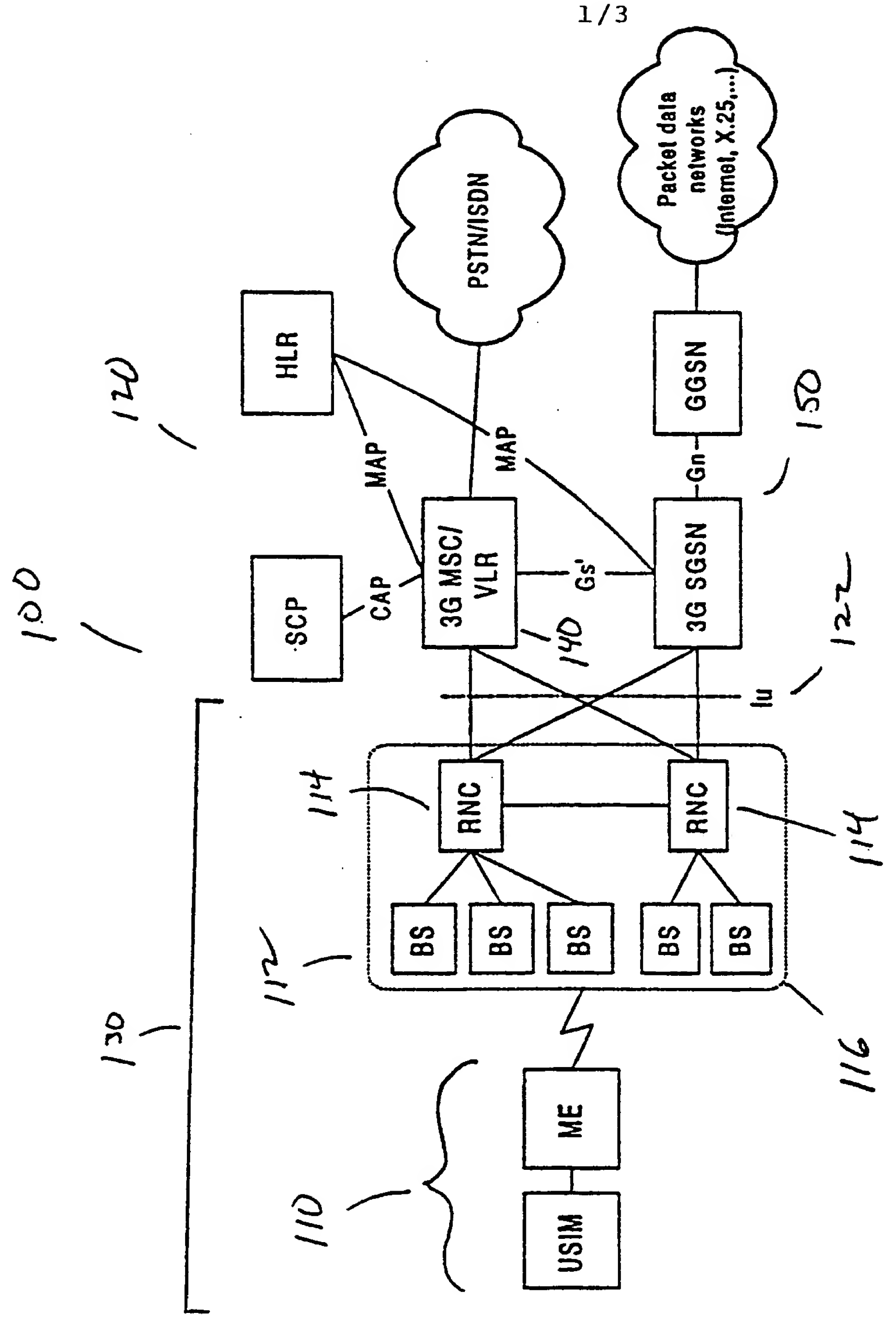
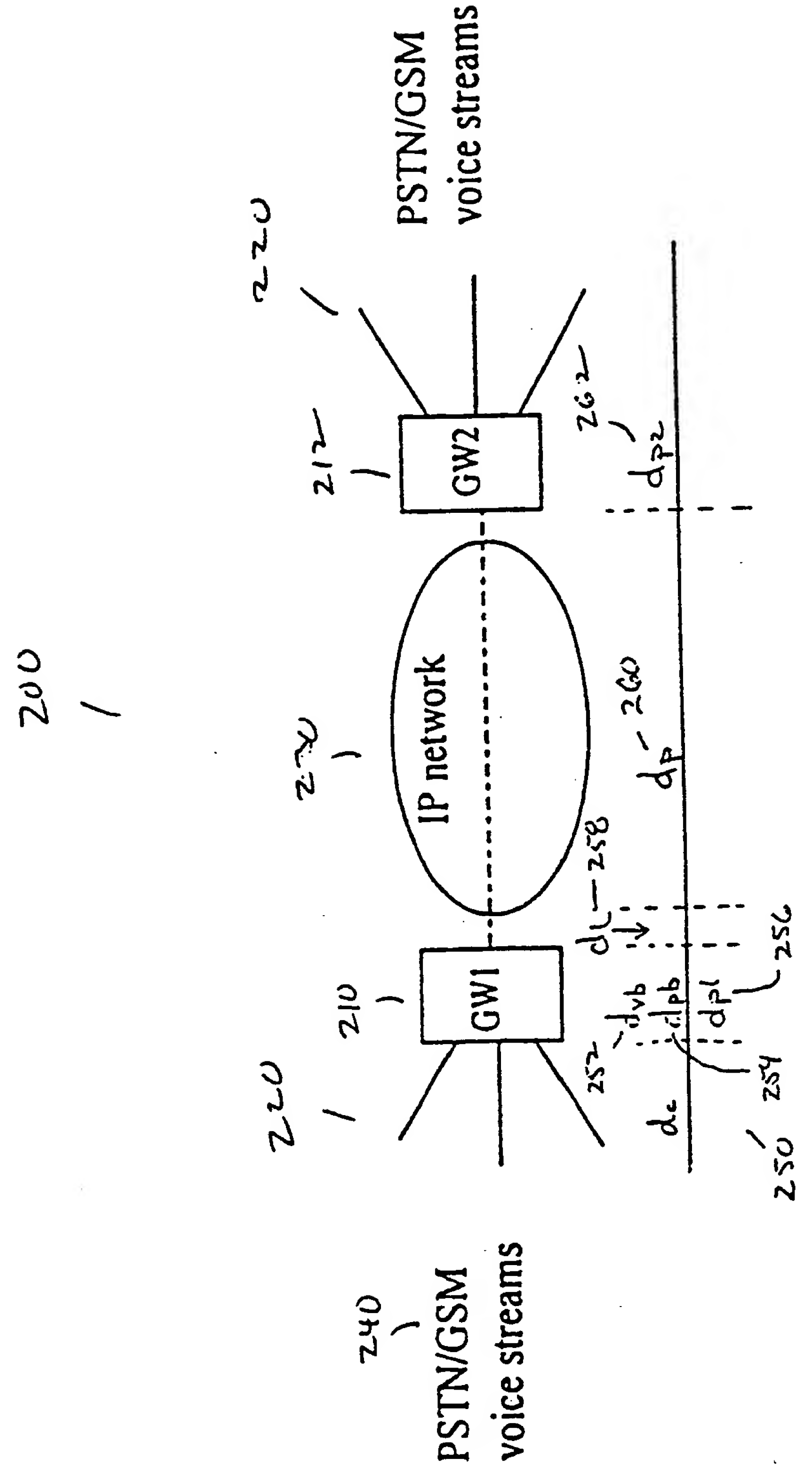

HLR
PSTN/ISDN
Packet data networks (Internet, X.25,...)
SCP
CAP
MAP
MAP
3G MSC/ VLR
Gs'
3G SGSN
Gn
GGSN
Iu
RNC
RNC
BS
BS
BS
BS
BS
ME
USIM
100
120
122
130
140
150
150
110
112
114
114
116

Fig. 1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IP network
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256
GW1
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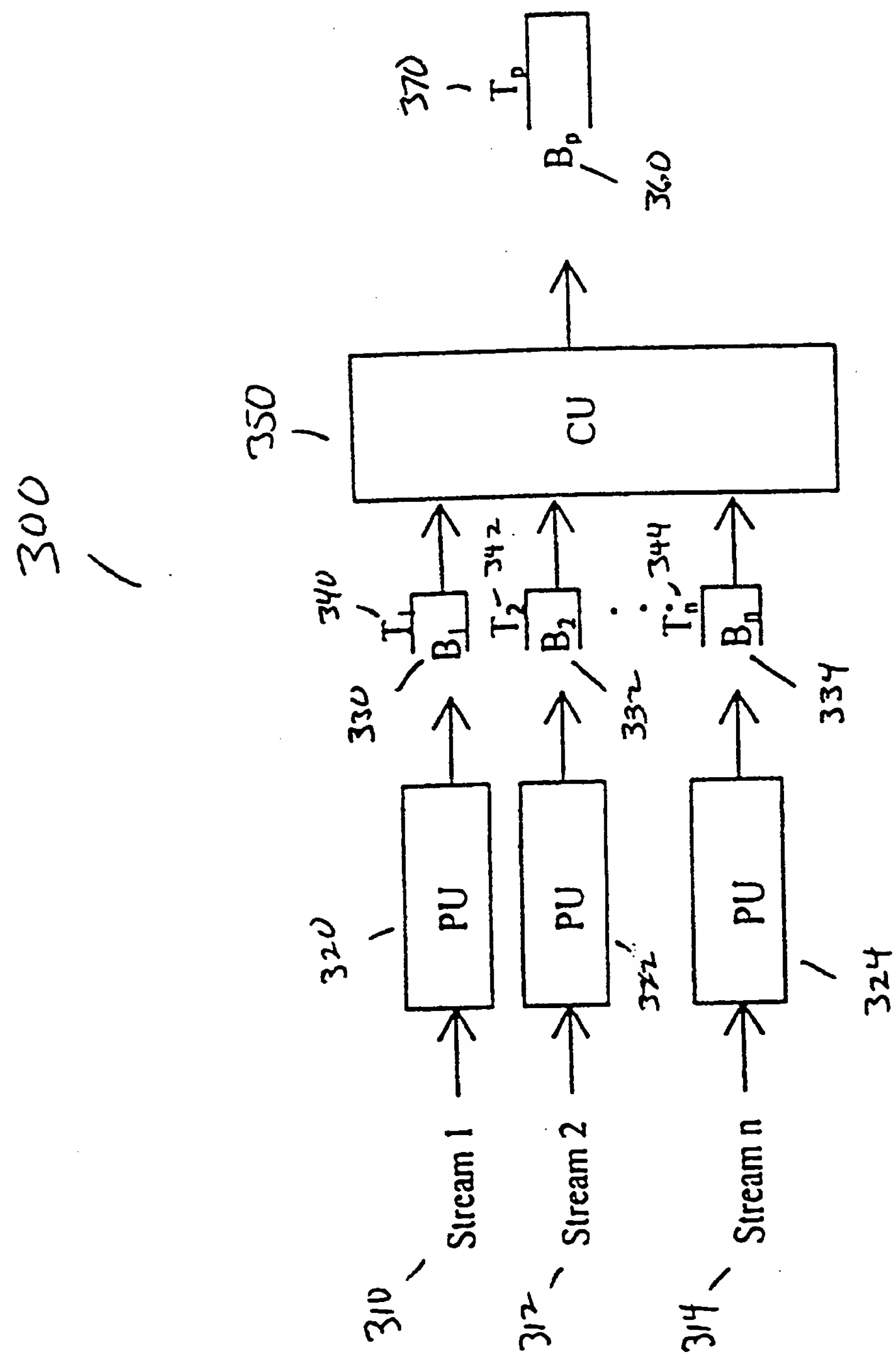
$d_{pl}$
252
254
$d_c$
220
250
240
PSTN/GSM voice streams

Fig. 2


300
310
Stream 1
312
Stream 2
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Stream n
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PU
322
PU
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PU
330
$B_1$
$T_1$
340
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$B_2$
$T_2$
342
334
$B_n$
$T_n$
344
350
CU
360
$B_p$
$T_p$
370

Fig. 3

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(74) Agent: ROLNIK, Robert; Nokia, 6000 Connection Dr., Mail Drop 1-4-755, Irving, TX 75039 (US).

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**Published:**
— *With international search report.*

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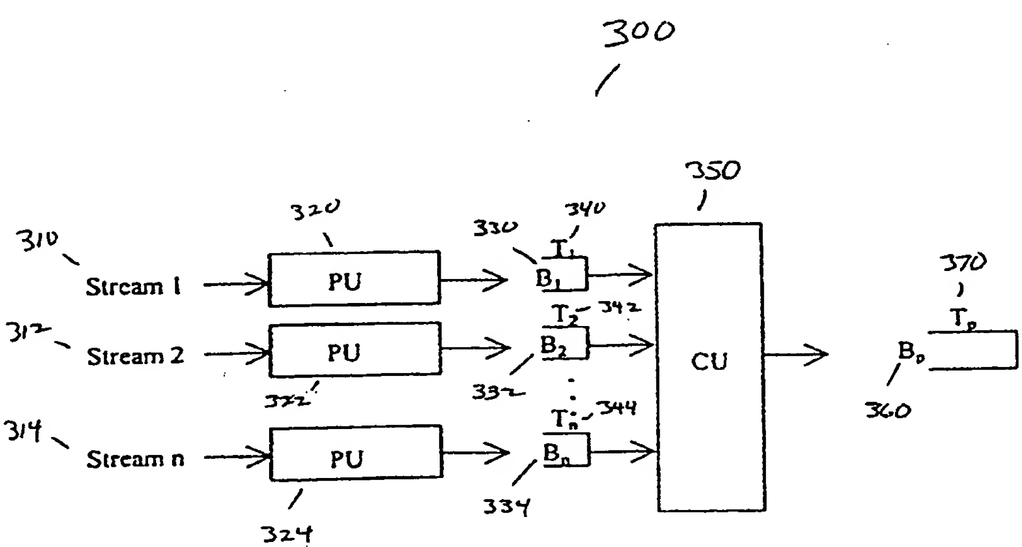
*For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.*

(54) Title: METHOD AND APPARATUS FOR PROVIDING EFFICIENT MULTIPLEXING BETWEEN GATEWAYS USING DYNAMIC TIMERS




(57) Abstract: A flexible mechanism employing timers within the gateways (300) to facilitate efficient multiplexing. Timers (340, 342, 344) are provided, wherein their value can be adjusted on a dynamic basis depending on factors such as network congestion that impact end-to-end delay. The extraction of data from the buffers (330, 332, 334) is triggered when either the timer (370) expires or when the accumulated data reaches a certain size. There are two ways one could set the timer values. In the first case, the network operator chooses the value based on the known approximation of end-to-end delay. In the second case, the multiplexing controller (350) has the capability to extract the network delay information from the RTCP reports.

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# INTERNATIONAL SEARCH REPORT

Intern al Application No
PCT/US 00/00607

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 7 H04L12/64

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category * | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| X | NAKAMURA H ET AL: "APPLYING ATM TO MOBILE INFRASTRUCTURE NETWORKS" ISS. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM),CA,TORONTO, PINNACLE GROUP, 21 September 1997 (1997-09-21), pages 73-80, XP000720509 * section 4.2 * figure 7 | 1 |
| X | US 5 541 926 A (SAITO TAKESHI ET AL) 30 July 1996 (1996-07-30) | 11,20 |
| A | column 6, line 50 -column 7, line 39; figure 3 | 10 |

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 13 July 2000 | 21/07/2000 |

| Name and mailing address of the ISA | Authorized officer |
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| European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016 | Gregori, S |

1

# INTERNATIONAL SEARCH REPORT

Interna  il Application No
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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

| Category ° | Citation of document, with indication,where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A | KITAMURA Y ET AL: "IMPLEMENTATION OF AAL2 FOR LOW BIT-RATE VOICE OVER ATM" ISS. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM),CA,TORONTO, PINNACLE GROUP, 21 September 1997 (1997-09-21), pages 271-276, XP000720533 page 274, column 2, line 18 -page 275, column 2, line 2; figures 6,8 | 1-28 |

1

# INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 00/00607

| Patent document cited in search report | Publication date | Patent family member(s) | Publication date |
|---|---|---|---|
| US 5541926 A | 30-07-1996 | JP 6169320 A | 14-06-1994 |